In the Claims

This listing of claims will replace all prior versions and listings of claims in the application:

1. (Previously Presented) A method for reducing noise in a sampled acoustic signal, comprising:

receiving a stream of sampled acoustic signals;

digitizing each sampled acoustic signal thereby forming digital samples;

selecting a fixed number of digital samples;

multiplying the digital samples by a windowing function;

computing the fast Fourier transform of the selected windowed digital samples to yield transformed windowed signals;

selecting half of the transformed windowed signals;

calculating a power estimate of the transformed windowed signals;

calculating a smoothed power estimate by smoothing the power estimate over time using the equation:

$$P^t(i) = (1-\alpha)\ P^{t-1}(i) + \alpha\ P(i)$$

where: $P^t(i)$ is the smoothed power estimate for a current time sample to be calculated for the i-th FFT point; $P^{t-1}(i)$ is the smoothed power estimate for an immediately prior time sample for the i-th FFT point; $P(i)$ is the calculated power estimate of the transformed windowed signals for the i-th FFT point; and α is an experimentally chosen predetermined value called the smoothing factor;

calculating a noise estimate;

calculating a gain function from the noise estimate and the smoothed power estimate;

calculating a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculating an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

adding the sampled speech signal to a portion of the speech signal of a previous frame.

2. (Original) The method of Claim 1, wherein the fixed number of samples is thirty-two.

3. (Original) The method of Claim 1, wherein the windowing function is a hanning window function.

9. (Previously Presented) A system for reducing noise in an acoustical signal comprising:

a sampler for obtaining discrete samples of the acoustical signal;

an analog to digital converter coupled to the sampler an operable to convert the analog discrete samples into a digitized sample;

a noise suppression circuit coupled to the analog to digital converter and operable to:

receive the digitized samples;

select a fixed number of digitized samples;

multiply the digitized samples by a windowing function;

compute the fast Fourier transform of the windowed digitized samples to yield transformed windowed signals;

select half of the transformed windowed signals;

calculate a power estimate of the transformed windowed signals;

calculate a smoothed power estimate by smoothing the power estimate over time using the equation:

$$P^t(i) = (1-\alpha)\ P^{t-1}(i) + \alpha\ P(i)$$

where: $P^t(i)$ is the smoothed power estimate for a current time sample to be calculated for the i-th FFT point; $P^{t-1}(i)$ is the smoothed power estimate for an immediately prior time sample for the i-th FFT point; $P(i)$ is the calculated power estimate of the transformed windowed signals for the i-th FFT point; and α is an experimentally chosen predetermined value called the smoothing factor;

calculate a noise estimate;

calculate a gain function from the noise estimate and the smoothed power estimate;

calculate a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculate an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

add the sampled speech signal to a portion of the speech signal of a previous frame.

10. (Original) The system of Claim 9, wherein the fixed number of samples is thirty-two.

11. (Original) The system of Claim 9, wherein the windowing function is a hanning window function.